TE405P/410P

User's Manual

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Safety Certification and Agency Approvals

Safety:

UL 60950-1:2003, First Edition
CSA C22.2 No. 60950-1-03 1st Ed. April 1, 2003
IEC 60950-1:2001 First Edition
EN 60950
AS/NZS 60950

> **Note:** Finland, Norway and Sweden require that equipment using this product must be located in a Restricted Access Location (RAL).

Telecom:

FCC Part 68, ANSI/ITA-968-A, Including Amendment A1 and A2
Industry Canada CS-03
AS/ACIF S016: 2001
AS/ACIF S038: 2001
TBR4 November 1995 as amended by TBR4/A1 December 1997
TBR12 December 1993
TBR13 January 1996
ICASA

Emissions:

47 CFR Part 15, Subpart B / 47 CFR Part 15, Subpart B, Class B
EN 55022:1998 Class B / EN 55022:1998 Class B Radiated and Conducted
EN 55024:1998 / IEC 61000

Immunity:

EN55024 ITE, EN61000

Federal Communications Commission Part 68

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the back of the TE405P/410P printed circuit board is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. If requested, this number must be provided to the telephone company.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA.

If the TE405P/410P causes harm to the telephone network, the telephone company may notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify you as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If you experience problems with the TE405P/410P, contact Digium, Inc. (+1.256.428.6161) for repair and/or warranty information. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

FCC Part 15

This device complies with part 15 of FCC rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) This device must accept any interference received, including interference that may cause undesired operation.

Industry Canada Compliance Information

Notice: The Industry Canada label applied to the product (identified by the Industry Canada logo or the "IC:" in front of the certification/ registration number) indicates that the Industry Canada technical specifications were met.

Notice: The Ringer Equivalence Number (REN) for this terminal equipment is supplied in the documentation or on the product labeling/ markings. The REN assigned to each terminal device indicates the maximum number of terminals that can be connected to a telephone interface. The termination on an interface may consist of any combination of devices so long as the sum of the RENs of all the devices does not exceed five (5).

Introduction to TE405P/410P Documentation

This manual contains product information for the TE405P/410P card. Be sure to refer to any supplementary documents or release notes that were shipped with your equipment. The manual is organized in the following manner:

Chapter/ Appendix	Title	Description
1	Overview	Identifies the type of card you received (TE405P or TE410P). This chapter covers applications and uses of the TE405P/410P in the real world.
2	Card Installation	Provides instructions for installing the card in your PC, acquiring correct drivers, and checking device compatibility.
3	Configuration	Provides examples for configuring dial plan options.
4	Troubleshooting	Explains resolutions to common problems and frequently asked questions pertaining to card installation and usage.
A	Pin Assignments	Lists the connectors and pin assignments.
B	Specifications	Details card specifications.
C	Asterisk Configuration Commands	Describes all available configuration commands and their option settings in the zapata.conf file.

D	Dial Plan Applications	Defines configuration features in the extensions.conf file.
E	Glossary and Acronyms	Defines terms related to this product.

Symbol Definitions

Caution statements indicate a condition where damage to the unit or its configuration could occur if operational procedures are not followed. To reduce the risk of damage or injury, follow all steps or procedures as instructed.

The ESD symbol indicates electrostatic sensitive devices. Observe precautions for handling devices. Wear a properly grounded electrostatic discharge (ESD) wrist strap while handling the device.

The Electrical Hazard Symbol indicates a possibility of electrical shock when operating this unit in certain situations. To reduce the risk of damage or injury, follow all steps or procedures as instructed.

Important Safety Instructions

User Cautions

Servicing.
Do not attempt to service this card unless specifically instructed to do so. Do not attempt to remove the card from your equipment while power is present. Refer servicing to qualified service personnel.

Water and Moisture.
Do not spill liquids on this unit. Do not operate this equipment in a wet environment.

Heat.
Do not operate or store this product near heat sources such as radiators, air ducts, areas subject to direct, intense sunlight, or other products that produce heat.

Static Electricity.
To reduce the risk of damaging the unit or your equipment, do not attempt to open the enclosure or gain access to areas where you are not instructed to do so. Refer servicing to qualified service personnel.

TABLE OF CONTENTS

Chapter 1
Overview 13
What is Asterisk? 18

Chapter 2
Card Installation 19
Unpacking the Card 19
Shipment Inspection 20
Identifying Communication Ports 20
T1/E1 Selection 23
Identifying Multiple Cards 23
Connecting Timing Cables 24
PCI Slot Compatibility 25
Hardware Installation 27
Software Installation 28

Chapter 3
Configuration 33
Configuring T1/E1 Lines 33
Testing Your configuration. 40

Chapter 4
Troubleshooting 43

Appendix A

Pin Assignments 49

Appendix B
Specifications 51

Appendix C
Asterisk Configuration Commands 53

Appendix D
Dial Plan Applications 61

Appendix E
Glossary and Acronyms 65

List of Figures

Figure 1: Sample Legacy Phone Application 15
Figure 2: Sample Channel Bank Application 16
Figure 3: Sample IP Phone Application 17
Figure 4: TE405P Card 21
Figure 5: TE410P Card 22
Figure 6: T1/E1 Jumpers 23
Figure 7: Ident Wheel 24
Figure 8: Timing Ports 25
Figure 9: Motherboard PCI Slots 26
Figure 10: Insert the Card 27
Figure 11: List PCI Slots Screen Capture 29
Figure 12: dmesg Screen Capture 40

List of Tables

Table A-1: RJ45 Telco Port Connector 49
Table B-2: Maximum Power Consumption 52

Chapter 1
Overview

The Digium TE405P/410P is a T1/E1 capable card for voice and data. It supports industry standard protocols, including Robbed Bit Signaling, E&M, Primary Rate ISDN (PRI), and several data modes (PPP, HDLC, Cisco HDLC and frame relay). It is capable of running in E1, T1, or J1 modes. It is also capable of DACSing channels from one span to another. The TE405P/410P is ideal for connecting phones to a channel bank, connecting to your T1/E1 switch, or connecting to a legacy PBX.

> **Note:** This manual is for use with the 5 volt TE405P and the 3.3 volt TE410P Quad T1/E1 PCI cards, and are identified collectively as the TE405P/410P card throughout this manual.

Designed to be fully compatible with existing software applications and integrate fully with Asterisk Open Source PBX/IVR platform, the TE405P/410P card allows many advanced call features.

Data Modes:

- Cisco HDLC
- HDLC
- PPP
- Multilink PPP
- Frame Relay

Voice Modes:

- PRI CPE and PRI NET
  - NI1
  - NI2
  - EuroISDN
  - 4ESS (AT&T)
  - 5ESS (Lucent)
  - DMS100

- E&M
  - Wink
  - Feature Group B
  - Feature Group D

- FXO and FXS
  - Ground Start
  - Loop Start
  - Loop Start with Disconnect Detect

The TE405P/410P card can be used to connect your Asterisk machine to the PSTN world, your channel bank, or even another PBX. This is accomplished via a T1/E1 interface. The card allows Asterisk PBX software to connect to your network, creating a professional telephony environment. Figure 2 shows an example of the card's primary application.

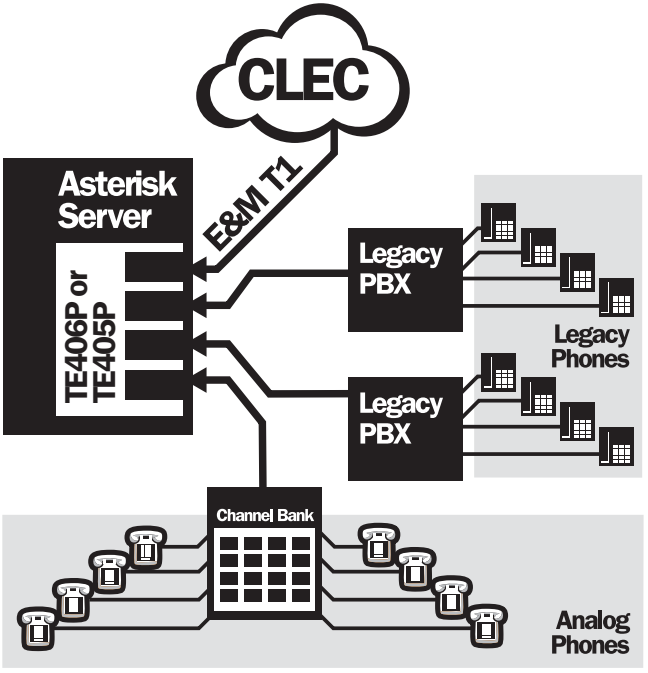


Figure 1: Sample Legacy Phone Application

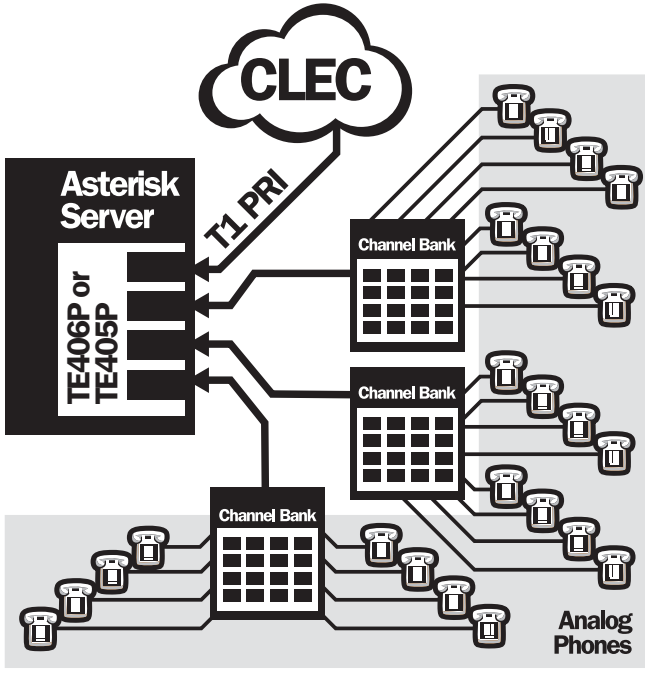


Figure 2: Sample Channel Bank Application

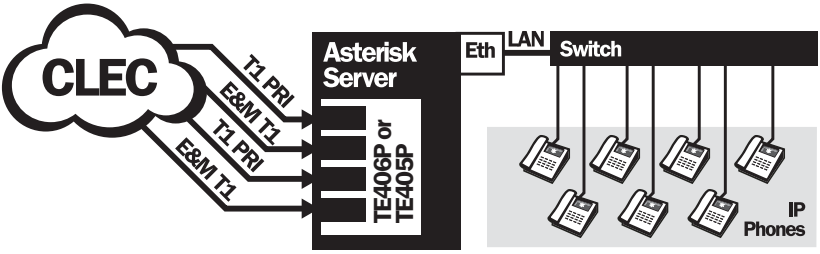


Figure 3: Sample IP Phone Application

What is Asterisk?

Asterisk is the first Open Source PBX. Since it runs on Linux, it inherits all of the power and stability of the operating system. The name Asterisk is derived from the all-inclusive “wildcard” symbol in UNIX, because it is opening a wide range of opportunities for developers worldwide to create solutions which would otherwise be cost-prohibitive.

Asterisk allows you to create a PBX solution that rivals the features and functionality of traditional telephony switches. Current PBX solutions are expensive and proprietary. International companies are discovering that Asterisk is cost effective, low maintenance, and flexible enough to handle all of their voice and data networking. Combined with Digium hardware and a common PC, anyone can replace an existing switch or complement a PBX by adding VoIP, voicemail, conferencing, and many other capabilities. Asterisk will integrate with most standards-based IP telephone handsets and software. Analog phones and ADSI-screen phones are also supported.

Chapter 2
Card Installation

This chapter provides the following information:

- **Unpacking the Card** on page 19
- **Shipment Inspection** on page 20
- **Identifying Communication Ports** on page 20
- **T1/E1 Selection** on page 23
- **Identifying Multiple Cards** on page 23
- **Connecting Timing Cables** on page 24
- **PCI Slot Compatibility** on page 25
- **Hardware Installation** on page 27
- **Software Installation** on page 28

Unpacking the Card

When you unpack your card, carefully inspect it for any damage that may have occurred in shipment. If damage is suspected, file a claim with the carrier and contact your reseller where the card was purchased, or Digium Technical Support (+1.256.428.6161). Keep the original shipping container to use for future shipment or proof of damage during shipment.

> **Note:** Only qualified service personnel should install the card. Users should not attempt to perform this function themselves.

Shipment Inspection

The following items are included in shipment of the TE405P/410P:

- TE405P or TE410P card.
- User's manual.
- Asterisk Standard Edition™ Linux Installation CD.
- T1 crossover cable, T1 cable, and loopback connector.

Identifying Communication Ports

The TE405P/410P card consists of four RJ45 ports and four status LEDs. The ports are used for connecting T1, E1, or J1 cables. See Figure 4 on page 21 and Figure 5 on page 22 to locate the ports and LEDs.

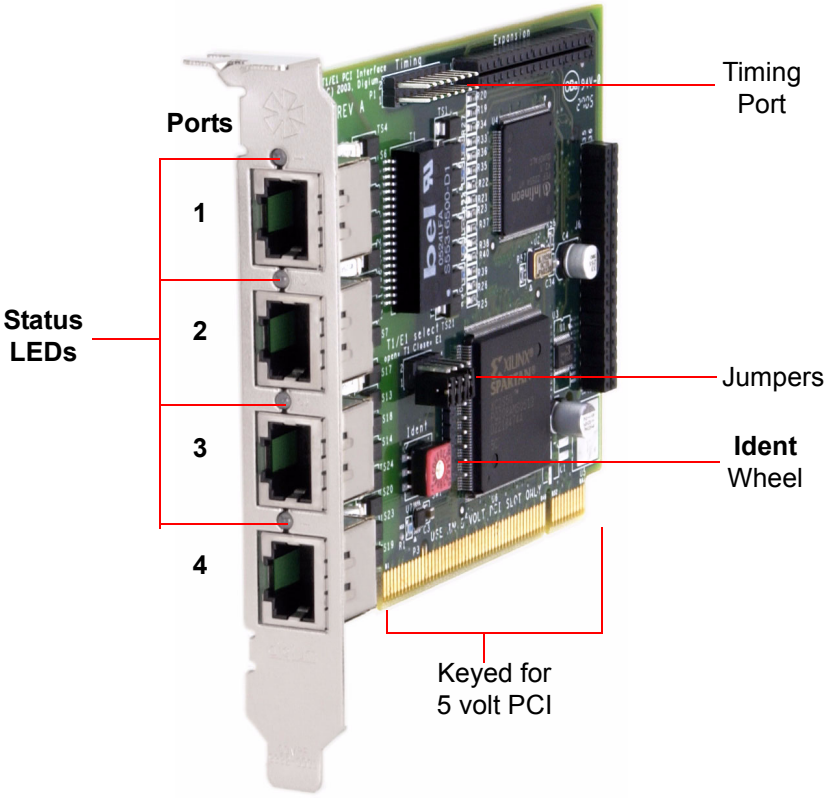


Figure 4: TE405P Card

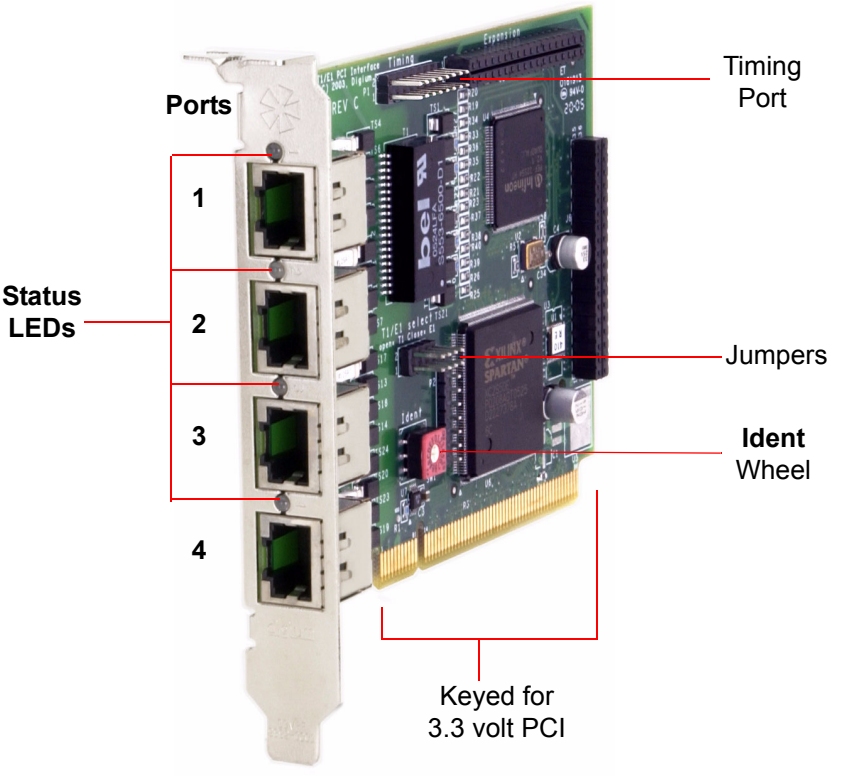


Figure 5: TE410P Card

T1/E1 Selection

The card includes a row of jumpers to select either T1 or E1 mode for the spans. The jumpers are shown in Figure 6. The T1/E1 mode, in most cases, is set at the distributor before shipment. You may want to check the setting to be certain they are set for your specific use.

Figure 6: T1/E1 Jumpers

Identifying Multiple Cards

If multiple TE405P/410P cards are installed in the same machine, then the **Ident** wheel can be used to control the order the cards are recognized. The click wheel switch with the word **Ident** printed above it can be set to a different number for each installed card. This number adjusts the order in which the driver recognizes the card. For example: set the first card **Ident** wheel to 0, set the second card **Ident** wheel to 1, and so on. The **Ident** wheel is shown in Figure 6.

Figure 7: Ident Wheel

Connecting Timing Cables

The timing port allows up to four TE405P/410P cards to share the same sync (timing) source from the T1 line provider, or provide a consistent sync source across multiple cards. This is a useful feature for data and fax modes and some voice applications to prevent corruption due to timing slips on the second, third or fourth TE405P/410P PC cards.

To utilize this feature, daisy-chain the P1 connector between each TE405P/410P card using the Digium 4-position timing cable. See Figure 8 on page 25 for an example. Enable this feature in the drivers using the **timingcable=1** switch when the drivers are loaded:

```
# modprobe wct4xxp timingcable=1
```

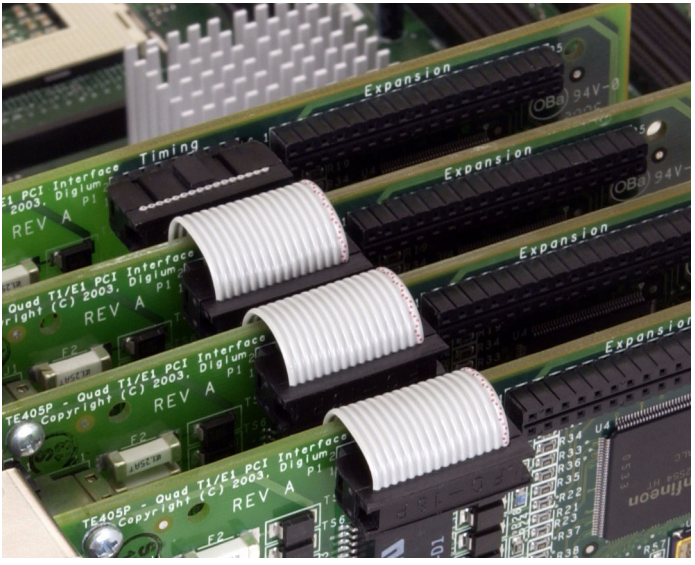


Figure 8: Timing Ports

Caution.
Only qualified service personnel should continue with hardware installation and configuration of the TE405P/410P card. Users should not attempt to perform these functions themselves.

PCI Slot Compatibility

Check the type of card you received to be sure it is compatible with your PCI slot. To determine which slot you have, identify it by comparing it to those shown in Figure 9 on page 26.

Slot Number:

0: AGP Pro Slot

1: 64-bit 5.0 volt PCI Slot

2: 64-bit 3.3 volt PCI Slot

3: 32-bit 5.0 volt PCI Slot

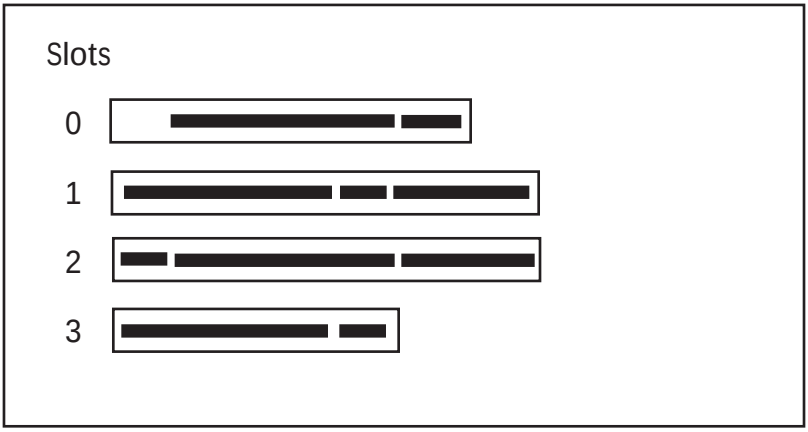


Figure 9: Motherboard PCI Slots

The TE405P card is a 32-bit 33MHz card keyed for 5.0 volt operation and works in any PCI 2.2 (or higher) compliant slot. This means that in the motherboard shown in Figure 9, the TE405P card will fit into Slots 1 and 3. The TE405P **will not** fit into Slot 2.

The TE410P card is a 32-bit 33MHz card keyed for 3.3 volt operation and works in any PCI 2.2 (or higher) compliant slot. This means that in the

motherboard shown in Figure 9, the TE410P card will only fit into Slot 2. The TE410P **will not** fit into Slots 1 or 3.

Hardware Installation

1. Now that you are acquainted with the card, power down your computer and unplug it from its power source.
2. Attach a static strap to your wrist and open the case.
3. Check the jumper setting to ensure it matches your equipment configuration. Setting the jumper with the strap **on** enables the ports for E1. Setting the jumper with the strap **off** enables the ports for T1.
4. Remove the bracket place holder and insert the card into the PCI slot. Be certain to place only a TE405P card into a 5 volt PCI slot or the TE410P card into a 3.3 volt PCI slot. See Figure 10.

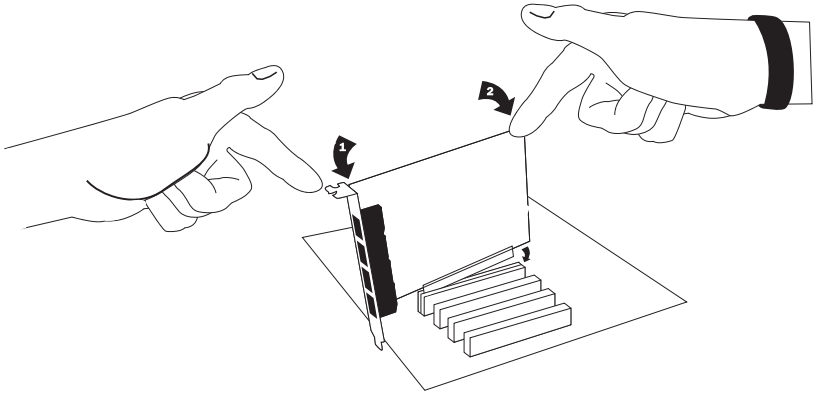


Figure 10: Insert the Card

5. Replace the cover to your computer.

6. Plug all T1 or E1 equipment cables into the RJ45 ports as needed.

Caution.
This unit must be connected to the Telecommunications Network in your country using an approved line cord, e.g.: for Australia use only line cords complying with ACA Technical Standard TS008.

Caution.
This unit must be connected only to the appropriate Telecommunications Network port (as approved for use in your specific country).

Software Installation

The card is only supported under Linux. Digium, Inc. recommends Asterisk Standard Edition, Debian, Fedora, and Red Hat, however, all other distributions are supported by Digium Technical Support. Digium hardware requires drivers and libraries that have not yet been integrated with the Linux kernel. You can obtain the source code from ftp.digium.com or svn.digium.com.

To install software for your TE405P/410P card, you will need:

- Full Linux kernel 2.4 (or later) source code
- Development libraries and headers for zlib, libedit, and openssl.

1. Check the PCI bus compatibility. Boot the computer into Linux. After the machine is loaded, log in and execute the following:

   ```
   # lspci
   ```

   You will see a screen similar to the one shown in Figure 11. Confirm your PCI bus compatibility by scanning for the information shown in the figure.

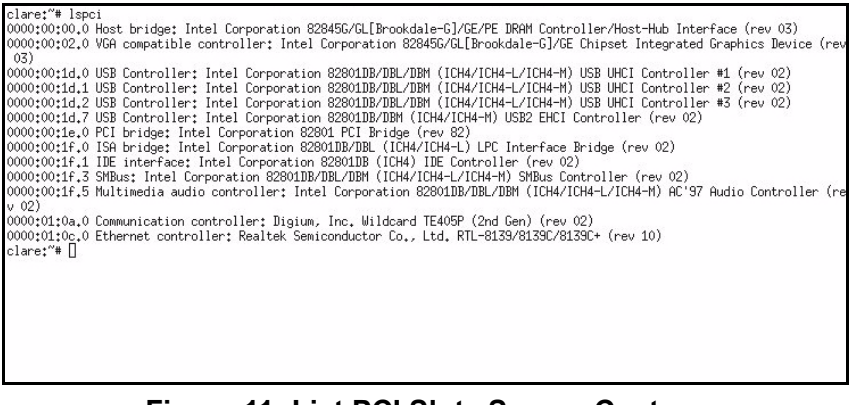

clare:~# lspci
0000:00:00.0 Host bridge: Intel Corporation 82845G/GL[Brookdale-G]/GE/PE DRAM Controller/Host-Hub Interface (rev 03)
0000:00:02.0 VGA compatible controller: Intel Corporation 82845G/GL[Brookdale-G]/GE Chipset Integrated Graphics Device (rev 03)
0000:00:1d.0 USB Controller: Intel Corporation 82801DB/DBL/DBM (ICH4/ICH4-L/ICH4-M) USB UHCI Controller #1 (rev 02)
0000:00:1d.1 USB Controller: Intel Corporation 82801DB/DBL/DBM (ICH4/ICH4-L/ICH4-M) USB UHCI Controller #2 (rev 02)
0000:00:1d.2 USB Controller: Intel Corporation 82801DB/DBL/DBM (ICH4/ICH4-L/ICH4-M) USB UHCI Controller #3 (rev 02)
0000:00:1d.7 USB Controller: Intel Corporation 82801DB/DBM (ICH4/ICH4-M) USB2 EHCI Controller (rev 02)
0000:00:1e.0 PCI bridge: Intel Corporation 82801 PCI Bridge (rev 82)
0000:00:1f.0 ISA bridge: Intel Corporation 82801DB/DBL (ICH4/ICH4-L) LPC Interface Bridge (rev 02)
0000:00:1f.1 IDE interface: Intel Corporation 82801DB (ICH4) IDE Controller (rev 02)
0000:00:1f.3 SMBus: Intel Corporation 82801DB/DBL/DBM (ICH4/ICH4-L/ICH4-M) SMBus Controller (rev 02)
0000:00:1f.5 Multimedia audio controller: Intel Corporation 82801DB/DBL/DBM (ICH4/ICH4-L/ICH4-M) AC'97 Audio Controller (rev 02)
0000:01:0a.0 Communication controller: Digium, Inc. Wildcard TE405P (2nd Gen) (rev 02)
0000:01:0c.0 Ethernet controller: Realtek Semiconductor Co., Ltd. RTL-8139/8139C/8139C+ (rev 10)
clare:~#

Figure 11: List PCI Slots Screen Capture

2. Check out the latest driver code and Asterisk software from ftp.digium.com or svn.digium.com. The following example is provided to check out the code from the SVN repository:

```
# cd /usr/src

# svn checkout http://svn.digium.com/svn/asterisk/
trunk asterisk
# svn checkout http://svn.digium.com/svn/zaptel/
trunk zaptel
# svn checkout http://svn.digium.com/svn/libpri/
trunk libpri
```

Commands to get the current snapshot from the release branch of SVN:

```
# svn checkout http://svn.digium.com/svn/asterisk/
branches/1.2 asterisk-1.2
# svn checkout http://svn.digium.com/svn/zaptel/
branches/1.2 zaptel-1.2
# svn checkout http://svn.digium.com/svn/libpri/
branches/1.2 libpri-1.2
```

This creates the following directories: zaptel, libpri, and Asterisk. Complete instructions for installing Asterisk is available at www.asterisk.org.

3. Install the zaptel drivers. Compiling the directories is straight forward. Change to each directory and type **`make install`**. Compile in this order: zaptel, libpri, and Asterisk.

   Type the following:

```
# cd /usr/src/zaptel
# make clean
# make install
```

   Note: If the build fails, it may be because you are missing one of the build dependencies, the kernel source, or development tools. Feel free to contact your reseller where the card was purchased, or call Digium Technical Support (+1.256.428.6161) for assistance.

4. Install libpri and Asterisk.

```
#cd /usr/src/libpri
#make install
#cd /usr/src/asterisk
#make
#make install
#make samples
```

   Note: This step installs sample configuration files which are necessary for Asterisk. You will make changes to these configuration files to set up your dial plan and customize Asterisk to suit your particular needs.

Chapter 3
Configuration

The TE405P/410P cards have a variety of configuration options. This chapter provides configurations for PRI, channel bank, E&M wink, and finally, data mode. These sample configurations are provided to assist you in familiarizing yourself with the flexibility of editing the configuration files to meet your specific needs. The list of possible configurations is too expansive to cover in this user manual.

Configuring T1/E1 Lines

1. Begin by opening the **zaptel.conf** file from the **/etc/asterisk/** directory.
2. Configure the SPAN Map.

   For each T1/E1 you are using you will need to define a span. The SPAN map includes defining the SPAN number, timing, line build out, framing and coding. Configuration details for each of these items is explained in this section.

```
span => NUMBER,TIMING,LINE BUILD OUT,FRAMING,CODING
```

Number:

This is the port the span is plugged into. Port 1 being the furthest span from the PCI bus. The port numbers are notated on the PCI bracket.

Timing:

Determines whether the card provides timing (0), takes timing (1), takes backup span timing (2), provides backup span timing (3), and so on. Only one span can be defined to take timing and it defines timing for the rest of the card spans.

Line Build Out

For most setups the line build out is 0.

0: 0 db (CSU) / 0-133 feet (DSX-1)

1: 133-266 feet (DSX-1)

2: 266-399 feet (DSX-1)

3: 399-533 feet (DSX-1)

4: 533-655 feet (DSX-1)

5: -7.5db (CSU)

6: -15db (CSU)

7: -22.5db (CSU)

Framing

T1 utilizes framing set for D4 (SF) or ESF. E1 utilizes CAS or CCS.

Coding

T1 coding can be AMI or B8ZS. E1 coding can be AMI or HDB3. E1 can also have the extra flag CRC4 at the end for CRC4 checking. Yellow flag can also be added at the end for transmitting a yellow alarm when all channels are in use.

The following is a typical setup for a telco in the US:

```
span => 1,1,0,esf,b8zs
```

In Europe:

```
span=>1,1,0,ccs,hdb3
```

First Example: Channel Bank

The Channel Bank in this example has 24 FXS ports. In this configuration, the **zaptel.conf** is set for the card to provide timing to the channel bank and **fxoks** is set for 24 stations.

Set **zapata.conf** to mirror the configuration with **`signalling=fxo_ks`** and define it for channels 1-24.

```
/etc/zaptel.conf:
span=1,0,0,esf,b8zs
fxoks=1-24


/etc/asterisk/zapata.conf:
group=1
context=channelbank
signalling=fxo_ks
channel=1-24
```

Second Example: E&M Line

In the E&M Line configuration, the **zaptel.conf** is set for the card to take timing from the telco on E&M with wink while **zapata.conf** mirrors the

configuration. However, Feat_D is a type of E&M with wink that accepts DID, but there are many E&M options; E&M_W, E&M, Feat_B, etc.

```
/etc/zaptel.conf:
span=1,1,o,esf,b8zs
e&m=1-24
/etc/asterisk/zapata.conf:
group=1
context=incoming
signalling=feat_d
channel=1-24
```

Third Example: PRI

By setting the card to take timing in **zaptel.conf**, you acquire 23 b channels and voice channels, with channel 24 as the data transport. For Asterisk, define PRI_CPE so it is the client side. Define the switch type you are connecting to as **national**. There are several options for the switch type including 5ESS, 4ESS, and NI1. You will then have 23 voice channels for Asterisk.

PRI T1

```
/etc/zaptel.conf:
span=1,1,0,esf,b8zs
bchan=1-23
dchan=24

/etc/asterisk/zapata.conf
group=1
signalling=pri_cpe
switchtype=national
context=incoming
channel=1-23
```

PRI E1

```
/etc/zaptel.conf:
span=1,1,0,ccs,hdb
bchan=1-15,17-31
dchan=16

/etc/asterisk/zapata.conf
group=1
signalling=pri_cpe
switchtype=euroisdn
context=incoming
channel=1-15,17-31
```

Fourth Example: Data Mode

Data mode is a little different than the other options. The **zaptel.conf** is configured as follows:

```
/etc/zaptel.conf
span=1,0,0,esf,b8zs
nethdlc=1-24
```

1. Instructions for Cisco HDLC:
   Compile kernel with HDLC support:

   Note: We suggest that you use either a Kernel version of 2.4.20 or less, or a Kernel of 2.6.8 or greater. The HDLC implementation in the interval kernels is in a state of too much flux. The following data modes are described in this section:
   - WAN Interfaces Support
   - Generic HDLC Layer
   - Cisco HDLC support

2. Rebuild and reboot into your kernel.
3. Uncomment the following line in **zconfig.h** of the Zaptel package:

   ```
   #define CONFIG_ZAPATA_NET
   ```

   If you are using a kernel prior to 2.4.19, also uncomment this line:

   ```
   #define CONFIG_OLD_HDLC_API
   ```

Rebuild Zaptel including the creation of the **SetHDLC** utility:

```
make sethdlc-new;use "make sethdlc" for
;kernels 2.4.19 and prior
make install
```

4. Load and configure your driver:

```
modprobe wct4xxp
ztcfg
```

5. Use **sethdlc** to bring up the interface:

```
sethdlc hdlc0 cisco
```

-or- for old style (make **sethdlc** instead of **sethdlc-new**) use:

```
sethdlc hdlc0 mode cisco
```

6. Assign the interface an address:

```
ifconfig hdlc0 192.168.0.1 netmask 255.255.255.0
```

7. The interface may be addressed as any other networking interface (i.e., eth0) in Linux.

Testing Your configuration.

1. Load Zaptel drivers into the kernel using the program **modprobe**. The appropriate driver for the TE405P/410P card is **wct4xxp**. Users in all countries except Australia should use the following modprobe command:

   # modprobe wct4xxp

   ztcfg -vv

   dmesg screen import

```
drivers/usb/input/hid-core.c: v2.0:USB HID core driver
usb 1-2: USB disconnect, address 2
usb 1-2.1: USB disconnect, address 3

Stopped TE4XXP, Turned off DMA
ACPI: PCI interrupt 0000:00:0c.0[A] -> GSI 10 (level, low) -> IRQ 10
Found TE4XXP at base address f4001400, remapped to c8883400
TE4XXP version c01a0164, burst OFF, slip debug: OFF
FALC version: 00000005, Board ID: 00
Reg 0: 0x01fff400
Reg 1: 0x01fff000
Reg 2: 0xffffffff
Reg 3: 0x00000000
Reg 4: 0x0000ff00
Reg 5: 0x00000000
Reg 6: 0xc01a0164
Reg 7: 0x00001000
Reg 8: 0x00000000
Reg 9: 0x00ff00ff
Reg 10: 0x00000000
TE4XXP: Launching card: 0
TE4XXP: Setting up global serial parameters
Found a Wildcard: Wildcard TE405P (2nd Gen)
herbie:~#
```

Figure 12: dmesg Screen Capture

2. Run **zttool** from the command line and see if the span turns green for each span you have connected.

```
zttool
```

3. Execute the following Asterisk command to see if the span came up successfully.

```
asterisk
asterisk -vvvr
```

Note: More detailed information is provided on troubleshooting in the Asterisk User's Manual. A copy may be requested by contacting Digium Technical Support (+1.256.428.6161) or visiting the website at www.digium.com. You may also contact your distributor or reseller where the card was purchased for assistance.

Chapter 4
Troubleshooting

This chapter provides frequently asked questions as identified from Digium Technical Support and possible resolutions. Multiple resources are available to obtain more information about Asterisk and Digium products. These resources are listed on page 47.

What do the LED colors indicate?

- Green - Card is in-sync with the far end.
- Yellow - Card is synchronizing.
- Red - Card is not seeing far end, circuit is not up, or cable is bad.

I can't receive DID calls even though I have it enabled in extensions.conf.

Your telco might be sending calls with a method you are not expecting.

1. Check the method being used by attempting the following in your line context:
   ```
   _x.,1,noop(My DID Matches as ${EXTEN}
   ```

2. Then type **`reload`** in the Asterisk console and call in. You should see the DID come in on your T1/E1 line.

My D Channel seems to go up and down.

Check to be sure you have set your timing parameters correctly. Also check the common causes of problems for a T1. See the **Common Fixes for all cards**, page 46.

I have trouble dialing out. It seems that one type of dialing works (local, long distance, international) but another does not.

Check your **pridialplan** variable and be sure that you are dialing using the method your telco is expecting.

I am having trouble receiving DID information over E&M.

Try the other types of E&M (featd, featb, etc.) to match the method your telco is using to stream information.

I am having issues with my PRI. How can I see the messages coming across my D channel?

Enter the following command:

```
PRI debug span X
```

where x is the port from which you are connected. This command will show you the PRI messages coming across your D channel for that message.

I am still having problems and the telco tells me it is my equipment.

The first thing to do in this situation is to test your equipment.

1. Plug in a loopback cable. (A loopback cable is a cable that has pin 1 going to pin 4 and pin 2 going to pin 5.) Plug the cable into the span and wait for its LED to turn green.

2. Stop Asterisk and edit **zaptel.conf** by removing the lines defined for your card and replacing them with the following:

   ```
   span=>1,0,0,esf,b8zs
   clear=1-24
   ```

   Or if you have an E1 span:

   ```
   span=> 1,0,0,ccs,hdb3
   clear=1-31
   ```

3. Navigate to your zaptel source directory and type:

   ```
   make tests
   ```

   Followed by:

   ```
   ./patlooptest /dev/zap/1 60
   ```

   The first argument in the patlooptest command is the device for the channel number you want to test. You should always test the first channel of a span. The second argument is the duration in seconds to run the test.

This runs a pattern looptest for 60 seconds. If you receive any failures, it is possible you have a bad card and will need to call Digium Technical Support (+1.256.428.6161)

Common Fixes for all cards

1. Check for shared interrupts by entering the following:

   ```
   cat /proc/interrupts
   ```

   or

   ```
   lspci -vb
   ```

   If a conflict exists, try moving the card to another PCI slot.

2. Check to see if X windows is running by entering the following:

   ```
   ps aux|grep X
   ```

   If X windows is running, stop the application since it may cause a conflict with Asterisk.

3. Check to see if your hard drives are running with DMA levels set. Perform an **hdparm** on your hard drive interface.
   If you are still having problems contact your reseller where the card was purchased, or Digium Technical Support (+1.256.428.6161).

How can I enable more features?

To view all of the options available to add to your dial plan, type the following command from within Asterisk:

```
show applications
```

Digium also offers services to help configure and add features you might need. Contact Digium Technical Support (+1.256.428.6161) for more information.

Where can I ask even more questions?

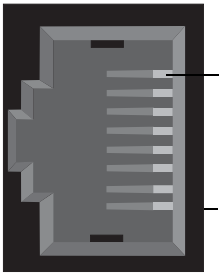
There are several places to inquire for more information about Asterisk Digium products:

1. Digium Technical Support (+1.256.428.6161) is available 7am-7pm Central Time, Monday - Friday.
2. Asterisk forums (forums.digium.com).
3. Asterisk users mailing list (asterisk.org/lists.digium.com).
4. IRC channel **#asterisk** on (irc.freenode.net).

Appendix A
Pin Assignments

All four ports on the TE405P/410P bracket are 8-pin RJ45 ports. The pin assignments are identified in Table A-1.

Table A-1: RJ45 Telco Port Connector

	Pin	Description
	1	Rx
	2	Rx
	3	Not used
	4	Tx
	5	Tx
	6	Not used
	7	Not used
	8	Not used

Appendix B
Specifications

This appendix provides specifications, required environmental conditions, and maximum power consumption for the TE405P/ 410P card.

Physical.

Size: 5” × 3.75” × 0.63” (12.7 x 9.53 x 1.6 cm)
PCB size, does not include the PCI bracket
Weight: 3.5 oz (100gm)

Interfaces.

Local Loop Access: E1, T1, J1, PRI; RJ45
PCI Bus: 3.3V or 5V bus slot, half-length slot minimum size, 33MHz minimum bus speed, compliant with PCI 2.2 or greater.

Environment.

Temperature: 0 to 50° C (32 to 122° F) operation
-20 to 65° C (4 to 149° F) storage
Humidity: 10 to 90% non-condensing

Hardware and Software Requirements.
800-Mhz Pentium III or better
64MB RAM
Available PCI Slot (as described previously)

Table B-2: Maximum Power Consumption

Model	Power
TE405P 3.3V 5V	 2 Watts 0 Watts
TE410P 3.3V 5V	 2 Watts Unused

Appendix C
Asterisk Configuration Commands

This appendix provides the configuration options available in the zapata.conf file which is created during the initial installation of Asterisk. To change any of the following features, edit the zapata.conf file with your text editor, saving the file upon completion. The following lists the command, description, and the available options:

adsi
ADSI (Analog Display Services Interface) can be enabled on a per-channel basis if you have (or may have) ADSI compatible CPE equipment.
yes / no

busycount
If enabled, it is also possible to specify how many busy tones to wait before hanging up.
positive integer

busydetect
On trunk interfaces (FXS) and E&M interfaces (E&M, Wink, Feature Group D, etc.), it can be useful to perform busy detection either in an effort to detect hangup or for detecting busy signals.
yes / no

cadence

You can define your own custom ring cadences, up to 8 pairs. If the silence is negative, it indicates where the callerid spill is to be placed. Also, if you define any custom cadences, the default cadences will be turned off.

Syntax is: **`cadence=ring,silence [,ring,silence[...]]`**

The default is based on the country code.

callerid

CallerID can be set to **asreceived** *or a specific number if you want to override it. Note that* **asreceived** *only applies to trunk interfaces.*

asreceived or a specific number

callgroup, pickupgroup

*Supports ring groups (a.k.a. call groups) and pickup groups. If a phone is ringing and it is a member of a group which is one of your pickup groups, then you can answer it by picking up and dialing *8#.*

For simple offices, set these both to the same positive integer.

callprogress, progzone

On trunk interfaces (FXS), it can be useful to attempt to follow the progress of a call through ringing, busy, and answering. If enabled, call progress attempts to determine answer, busy, and ringing on phone lines. This feature is HIGHLY EXPERIMENTAL and can easily detect false answers, so DO NOT rely on it being accurate.

Few zones are supported at the time of this writing, but may be selected with progzone, yes / no, and two letter country code for progzone.

callreturn

*Support call return (*69).*

yes / no

callwaiting
Enable call waiting on FXO lines.
yes / no

callwaitingcallerid
Support caller ID on call waiting.
yes / no

canpark
Allow call parking.
yes / no

cidsignalling
Type of caller ID signalling in use.
bell =bell202 (US)
v23 = v23 (UK)
dtmf = DTMF (Denmark, Sweden, & Netherlands)

cidstart
Identifies what signals the start of caller ID.
ring = a ring signals the start
polarity = polarity reversal signals the start

echocancel
Enable echo cancellation.
yes / no, or a power of two from 32 to 256 (if you want to actually set the number of cancellation taps)

echocancelwhenbridged
Generally, it is not necessary (and in fact undesirable) to echo cancel when the circuit path is entirely TDM. You may, however, reverse this behavior by enabling the echo cancel during pure TDM bridging.
yes / no

echotraining
In some cases, the echo canceller doesn't train quickly enough and there is echo at the beginning of the call. Enabling echo training will cause Asterisk to briefly mute the channel, send an impulse, and use the impulse response to pre-train the echo canceller so it can start out with a much closer idea of the actual echo.
Value may be yes, no, or a number of milliseconds (positive int) to delay before training.
(default = 400)

faxdetect
Upon fax detection, routes fax to a fax extension.
Both, incoming, outgoing, or no.

group
Logical groups can be assigned to allow outgoing rollover.
Groups range from 0 to 63, and multiple groups can be specified.

hanguponpolarityswitch
In some countries, a polarity reversal is used to signal the disconnect of a phone line. If this option is enabled, the call will be considered disconnected on a polarity reversal.
yes / no

hidecallerid
*Hide outgoing caller ID. Override with *67 or *82.*
yes / no

immediate
Specify whether the channel should be answered immediately or if the simple switch should provide dialtone, read digits, etc.
yes / no

jitterbuffers
Configure jitter buffers in zapata.
(each one is 20ms, default is 4) positive integer

mailbox
Supports stutter dialtone.
If a mailbox is specified without a voicemail context, then when message is received, taking the phone off hook will cause a stutter dialtone instead of a normal dialtone. If a mailbox is specified with a voicemail context, the same will result if a message is received in the specified voicemail context.
mailbox ex. 1234, or 1234@other.

musiconhold
Select which class of music to use for music on hold. If not specified then the default will be used. The music class is defined in musiconhold.conf file.
default, loud, random

pulsedial
Use pulse dial instead of DTMF for FXO (FXS signalled) devices.
yes / no

relaxdtmf
If you have trouble with DTMF detection, you can relax the DTMF detection parameters. Relaxing them may make the DTMF detector more likely to have talk off where DTMF is detected incorrectly.
yes / no

rxgain
Receive gain.
Positive or negative double, measured in dB. Digium recommends a range of -5 to 5.

sendcalleridafter
Some countries have ring tones with a set of cadences which differ from the default (e.g. UK uses ring-ring). This requires the callerid to be set with a delay, and not right after the first ring (the default).
Positive integer

threewaycalling
Support three-way calling.
yes / no

transfer
Support flash-hook call transfer (requires three way calling). Also enables call parking (overrides the **canpark** *parameter).*
yes / no

txgain
Transmit gain.
Positive or negative double, measured in dB. Digium recommends a range of -5 to 5.

usecallerid
Whether or not to use caller ID.
yes / no

usedistinctiveringdetection
Indicates whether or not to allow distinctive ring detection on FXO lines.
yes / no

Appendix D
Dial Plan Applications

This appendix details the applications available for configuring the dial plan. The dial plan is customized in the extensions.conf file which is created during the initial installation of Asterisk. This file controls how all calls are handled and routed. To change any of the following options, edit the extensions.conf file, saving the file upon completion. The following defines each feature:

AgentCallbackLogin
Call agent callback login

AgentLogin
Call agent login

AGI
Executes an AGI compliant application

Answer
Answer a channel if ringing

Authenticate
Authenticate a user

BackGround
Play a file while awaiting extension

BackgroundDetect
Background a file with talk detect

Busy
Indicate busy condition and stop

Dial
Place a call and connect to the current channel

Directory
Provide directory of voicemail extensions

DISA
DISA (Direct Inward System Access)

Goto
Go to a particular priority, extension, or context

GotoIf
Conditional goto

GotoIfTime
Conditional goto on current time

Hangup
Unconditional hangup

Macro
Macro Implementation

Monitor
Monitor a channel

MusicOnHold
Play music on hold indefinitely

Playback
Play a file

Queue
Queue a call for a call queue

Record
Record to a file

SetCallerID
Set CallerID

SetCallerPres
Set CallerID presentation

SetCIDName
Set CallerID name

SetCIDNum
Set CallerID number

VoiceMail
Leave a voicemail message

VoiceMailMain
Enter voicemail system

Zapateller
Block telemarketers with SIT

Appendix E
Glossary and Acronyms

ANSI *American National Standards Institute*

An organization which proposes and establishes standards for international communications.

asynchronous

Not synchronized; not timed to an outside clock source. Transmission is controlled by start bits at the beginning and stop bits at the end of each character. Asynchronous communications are often found in internet access and remote office applications.

attenuation

The dissipation of a transmitted signal's power as it travels over a wire.

bandwidth

The capacity to carry traffic. Higher bandwidth indicates the ability to transfer more data in a given time period.

bit

The smallest element of information in a digital system. A bit can be either a zero or a one.

bps *bits per second*

A measurement of transmission speed across a data connection.

broadband

Broadband transmission shares the bandwidth of a particular medium (copper or fiber optic) to integrate multiple signals. The channels take up different frequencies on the cable, integrating voice, data, and video over one line.

channel

A generic term for an individual data stream. Service providers can use multiplexing techniques to transmit multiple channels over a common medium.

Cat5

Category of Performance for wiring and cabling. Cat 5 cabling support applications up to 100 MHz.

Cat5E

Category of Performance for wiring and cabling. Category 5 Enhanced wiring supports signal rates up to 100 MHz but adheres to stricter quality specifications.

CLEC *competitive local exchange carrier*

A term for telephone companies established after the Telecommunications Act of 1996 deregulated the LECs. CLECs compete with ILECs to offer local service. See also *LEC* and *ILEC*.

CO *central office*

The CO houses local switching equipment. All local access lines in a particular geographic area terminate at this facility (which is usually owned and operated by an ILEC).

CPE *customer premises equipment*

Terminal equipment which is connected to the telecommunications network and which resides within the home or office of the customer. This includes telephones, modems, terminals, routers, and television set-top boxes.

DS0 *Digital Signal, Level 0*

A voice grade channel of 64 Kbps. The worldwide standard speed for digitizing voice conversation using PCM (Pulse Code Modulation).

DS1 *Digital Signal, Level 1*

1.544 Mbps in North America (T1) and Japan (J1) -up to 24 voice channels (DS0s), 2.048 Mbps in Europe (E1) - up to 32 voice channels (DS0s). DS1/T1/E1 lines are part of the PSTN.

DS3 *Digital Signal, Level 3*

T3 in North America and Japan, E3 in Europe. Up to 672 voice channels (DS0s). DS3/T3/E3 lines are not part of the PSTN

DTMF *Dual Tone Multi-Frequency*

Push-button or touch tone dialing.

E1

The European equivalent of North American T1, transmits data at 2.048 Mbps, up to 32 voice channels (DS0s).

E3

The European equivalent of North American T3, transmits data at 34.368 Mbps, up to 512 voice channels (DS0s). Equivalent to 16 E1 lines.

EMI *Electromagnetic Interference*

Unwanted electrical noise present on a power line

full duplex

Data transmission in two directions simultaneously.

FXO *Foreign Exchange Office*

Receives the ringing voltage from an FXS device. Outside lines are connected to the FXO port on your TE405P/410P card.

FXS *Foreign Exchange Station*

Initiates and sends ringing voltage. Phones are connected to the FXS ports on the TE405P/410P card.

G.711

The International Telecommunications Union recommendation for an algorithm designed to transmit and receive mulaw PCM voice and A-law at digital bit rate 64 Kbps. This algorithm is used for digital telephone sets on digital PBX.

G.729

An International Telecommunications Union standard for voice algorithm.

H.323

An International Telecommunications Union standard for multimedia communications over packet-based networks.

IAX *Inter-Asterisk eXchange*

A VoIP protocol used by Asterisk. It is used to enable VoIP connections between Asterisk servers, and between servers and clients that also use the IAX protocol.

iLBC *internet Low Bitrate Codec*

A free speech codec used for voice over IP. It is designed for narrow band speech with a payload bitrate of 13.33 kbps (frame length = 30ms) and 15.2 kbps (frame length = 20 ms).

ILEC *incumbent local exchange carrier*

The LECs that were the original carriers in the market prior to the entry of competition and therefore have the dominant position in the market.

interface

A point of contact between two systems, networks, or devices.

ISO *International Standards Organization*

LED *light-emitting diode*

Linux

A robust, feature-packed open source operating system based on Unix that remains freely available on the internet. It boasts dependability and offers a wide range of compatibility with hardware and software. Asterisk is supported exclusively on Linux.

loopback

A state in which the transmit signal is reversed back as the receive signal, typically by a far end network element.

MGCP *Media Gateway Control Protocol*

multiplexing

Transmitting multiple signals over a single line or channel. FDM (frequency division multiplexing) and TDM (time division multiplexing) are the two most common methods. FDM separates signals by dividing the data onto different carrier frequencies, and TDM separates signals by interleaving bits one after the other.

MUX *multiplexer*

A device which transmits multiple signals over a single communications line or channel. See multiplexing.

PBX *private branch exchange*

A smaller version of a phone company's large central switching office. Example: Asterisk.

PCI *peripheral component interconnect*

A standard bus used in most computers to connect peripheral devices.

POP *point of presence*

The physical connection point between a network and a telephone network. A POP is usually a network node serving as the equivalent of a CO to a network service provider or an interexchange carrier.

POTS *plain old telephone service*

Standard phone service over the public switched telephone network (PSTN). This service provides analog bandwidth of less than 4 kHz.

PPP *point-to-point protocol*

Type of communications link that connects a single device to another single device, such as a remote terminal to a host computer.

PSTN *public switched telephone network*

A communications network which uses telephones to establish connections between two points. Also referred to as the dial network.

QoS *quality of service*

A measure of telephone service, as specified by the Public Service Commission.

RJ11

A six-pin jack typically used for connecting telephones, modems, and fax machines in residential and business settings to PBX or the local telephone CO.

SIP *Session Initiation Protocol*

An IETF standard for setting up sessions between one or more clients. It is currently the leading signaling protocol for Voice over IP, gradually replacing H.323.

T1

A dedicated digital carrier facility which transmits up to 24 voice channels (DS0s) and transmits data at 1.544 Mbps. Commonly used to carry traffic to and from private business networks and ISPs.

T3

A dedicated digital carrier facility which consists of 28 T1 lines and transmits data at 44.736 Mbps. Equivalent to 672 voice channels (DS0s).

TDM *time division multiplexer*

A device that supports simultaneous transmission of multiple data streams into a single high-speed data stream. TDM separates signals by interleaving bits one after the other.

telco

A generic name which refers to the telephone companies throughout the world, including RBOCs, LECs, and PTTs.

tip and ring

The standard termination on the two conductors of a telephone circuit; named after the physical appearance of the contact areas on the jack plug.

twisted pair

Two copper wires commonly used for telephony and data communications. The wires are wrapped loosely around each other to minimize radio frequency interference or interference from other pairs in the same bundle.

V *volts*

VoIP *Voice over IP*

Technology used for transmitting voice traffic over a data network using the Internet Protocol.

Zaptel (Zap)

Zapata Telephony Project dedicated to implementing a reasonable and affordable Computer Telephony platform into the world marketplace.